

Mediant 800 MSBG E-SBC, Mediant 1000 MSBG E-SBC and Mediant 3000 E-SBC Media Gateway

Configuration Note

Connecting AT&T's IP Flexible Reach - MIS/PNT/AVPN SIP Trunking Service

to

Microsoft® Lync Server 2010

Using

AudioCodes' Mediant 800 E-SBC, Mediant 1000 MSBG E-SBC and Mediant 3000 E-SBC Media Gateway

Document #: LTRT-38100



This document was published by the AudioCodes Interoperability group.

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Reader's Notes

Notice

This document describes the procedure for integrating the AT&T IP Flexible Reach-MIS/PNT/AVPN SIP Trunking service with Microsoft® Lync Server using the AudioCodes Mediant 800 MSBG-E-SBC, Mediant 1000 MSBG E-SBC and Mediant 3000 E-SBC Media Gateway.

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Date Published: June-12-2011

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Abbreviations and Terminology

Each abbreviation, unless widely used, is spelled out in full when first used.



Note: Throughout this guide, the term *E-SBC device* refers to AudioCodes' Mediant 800 MSBG E-SBC, Mediant 1000 MSBG E-SBC and the Mediant 3000 E-SBC Media Gateway.



Related Documentation

Manual Name
LTRT-26901_SIP_CPE_Release_Notes_Ver6.2.pdf
LTRT-52306_SIP_CPE_Product_Reference_Manual_Ver_6.2.pdf
LTRT-27001 Mediant 1000 MSBG User's Manual Ver 6.2.pdf
LTRT-40809 Mediant 1000 MSBG Installation Manual Ver 6.2.pdf
LTRT-89710 Mediant 3000 SIP User's Manual Ver 6.2.pdf
LTRT-94708 Mediant 3000 SIP-MGCP-MEGACO Installation Manual Ver 6.2.pdf

1 Introduction

This Configuration Guide describes a sample configuration for a network that uses the AudioCodes Mediant 800 MSBG E-SBC, Mediant 1000 MSBG E-SBC or the Mediant 3000 E-SBC Media Gateway to facilitate a connection between Microsoft Lync 2010 and AT&T's IP Flexible Reach MIS/PNT/AVPN SIP Trunking service for superior voice quality services.

The Mediant 800 MSBG E-SBC is a networking device that combines multiple service functions such as a Media Gateway, Session Border Controller (SBC), Data Router and Firewall, LAN switch, WAN access, Stand Alone Survivability (SAS) and an integrated general-purpose server.

The Mediant 1000 MSBG E-SBC is all-in-one multi-service access solution products for Service Providers (SME's) offering managed services and distributed Enterprises seeking integrated services. This multi-service business gateway is designed to provide converged Voice & Data services for business customers at wire speed, while maintaining SLA parameters for superior voice quality.

The Mediant 1000 MSBG E-SBC is based on AudioCodes' VolPerfect Media Gateway technology, combined with Enterprise class Session Border Controller, Data & Voice security elements, Data Routing, LAN Switching and WAN Access. These services allow smooth connectivity to cloud services, while providing protection to the end customer.

The Mediant 3000 E-SBC Media Gateway is a High Availability VoIP Gateway and Enterprise Class SBC for medium and large enterprises.



Note: The scope of this document does not cover security aspects for connecting the SIP Trunk to the Microsoft Lync environment. Security measures should be implemented in accordance with your organization's security policies. For basic security guidelines, see the 'AudioCodes Security Guidelines'.



Reader's Notes

2 **Testing Considerations**

Note the following special considerations for the AT&T test environment:

- Fax was not tested and is not supported.
- G.711 U-law is the only codec supported for this application.
- Music on hold does not work on IP Flexible Reach telephone numbers (TNs) that are served by the AT&T legacy SBC local PSTN footprint.
- Voice mail should work; however was not tested during the certification.
- Emergency 911/E911 Services Limitations and Restrictions Although AT&T provides 911/E911 calling capabilities, AT&T does not warrant or represent that the equipment and software (e.g., IP PBX) reviewed in this customer Configuration guide, will properly operate with AT&T IP Flexible Reach to complete 911/E911 calls; therefore, it is Customer's responsibility to ensure proper operation with its equipment/software vendor.
- The transfer calls were tested with REFER and AudioCodes terminates the messages and supports the call flow directly.

Transfer calls were tested with the Microsoft Lync environment pre-configured to send REFER messages towards the SIP trunk. As REFER messages were sent from the Microsoft Lync environment, the messages were processed by AudioCodes directly and not forwarded on to the AT&T IP Flexible Reach trunk. Direct handling of these messages is a proven AudioCodes feature and a proven interworking functionality between AudioCodes and Microsoft Lync 2010.

While AT&T IP Flexible Reach services support E911/911 calling capabilities under certain Calling Plans, there are circumstances when the E911/911 service may not be available, as stated in the Service Guide for AT&T IP Flexible Reach found at http://new.serviceguide.att.com <a hr



Reader's Notes

3 Scenario Overview

The configuration scenario described in this document includes the following setup:

- An Enterprise has a deployed Microsoft® Lync server 2010 in its private network for enhanced communication within the company.
- The enterprise decides to offer its employees Enterprise voice and to connect the company to the PSTN network using the Flexible Reach SIP Trunking service.

The setup requirements are as follows:

- While the Microsoft® Lync Server 2010 environment is located on the Enterprise's Local Area Network (LAN), the Flexible Reach SIP Trunks are located on the WAN.
- Microsoft® Lync Server 2010 works with the TCP transport type, while the Flexible Reach SIP trunk works on the SIP over UDP transport type.
- Both Microsoft® Lync Server 2010 and Flexible Reach SIP Trunk support the G.711-Ulaw coder type.
- Support for early media handling
- Support for call forwarding

The figure below illustrates an overview of the configuration scenario.



Figure 3-1: Scenario Overview



Reader's Notes

4 Configuring Microsoft Lync Server 2010

This section describes how to configure the Microsoft Lync Server 2010 to operate with the E-SBC device. This section describes the following procedures:

- Configuring the E-SBC device as an 'IP/PSTN Gateway'. See Section 4.1 on page 16.
- 2. Associating the 'IP/PSTN Gateway' with the Mediation Server. See Section 4.2 on page 21.
- **3.** Configuring a 'Route' to utilize the SIP trunk connected to the E-SBC device. See Section 4.3 on page 27.



Note: Dial Plans, Voice Policies, and PSTN usages are also necessary for enterprise voice deployment; however, they are beyond the scope of this document.

4.1 Configuring the AudioCodes E-SBC device as a 'IP/PSTN Gateway'

This section describes how to configure the E-SBC device as an IP/PSTN Gateway.



Note: The Microsoft Lync Topology Builder interface dialogs refer to the E-SBC device as an 'IP/PSTN gateway' or 'PSTN gateway'.

Help and Support

Windows Security

Run...

0

9

- To configure the E-SBC device as a IP/PSTN Gateway and associating it with the Mediation Server:
- 1. On the server where the Topology Builder is located, start the Microsoft Lync Server 2010 Topology Builder: Click Start, select All Programs, then select Lync Server Topology Builder.



) Microsoft SQL Server 2008

脂 Startup

Back

Start Search

Figure 4-1: Starting the Lync Server Topology Builder

The following screen is displayed:

Figure 4-2: Topology Builder Options

🔜 Topology Builder 🛛 🗙
Welcome to Topology Builder. Select the source of the Lync Server 2010 (RC) topology document.
Download Topology from existing deployment Retrieve a copy of the current topology from the Central Management Store database and save it as a local file. Use this option if you are editing an existing deployment.
Open Topology from a local file Open an existing Topology Builder file. Use this option if you have work in progress or if you have exported a topology from Planning Tool.
New Topology Create a blank topology and save it to a local file. Use this option for defining new deployments from scratch.
OK Cancel

 Choose 'Download Topology from the existing deployment and click OK. You are prompted to save the Topology which you have downloaded.

Figure 4-3: Save Topology

🌄 Save Topology As				×
Administr	ator 👻 Documents	- 🛃	Search	2
🌒 Organize 👻 📗 Views	👻 📑 New Folder			0
Favorite Links Desktop Computer Documents Pictures Nusic Recently Changed Searches Public	Name A	▼ Date modified ▼ 10/7/2010 5:53 PM 10/12/2010 10:5	Type TBXML File TBXML File	▼ Size ▼ Tac 101 KB 101 KB
Folders File <u>n</u> ame: Inter Save as type: Topol	op2.tbxml ogy Builder files (*.tbxml)			
			<u>S</u> ave	Cancel

3. Enter new **File Name** and **Save** – this action enables you to rollback from any changes you make during the installation.

The Topology Builder screen with the topology downloaded is displayed.

KLync Server 2010 (RC), Topology Builder			
Eile Action View Help			
Lync Server 2010 (RC)			Actions
🛨 🔃 Interop	SIP domain	•	Lync Server 2010 (RC)
	Default SIP domain	Orsw14 local	🔛 New Central Site
	Additional supported SIP	Not configured	Edit Properties
	domains:		New Topology
			Open Topology
	Simple URLs		Download Topology
			Save a copy of Topology
	Phone access URLs:	Active Simple URL	Publish Topology
		https://dialin.Ocsw14.local	Install Database
	Meeting URLs:	Active Simple URL SIP domain	Merge 2007 or 2007 R2 T
		https://meet.Ocsw14.local Ocsw14.local	Remove Deployment
	Administrative access URL:	Not configured	View >
			? Help
	Central Management Serve	er 🔺	-
	Central Management Server:	fe-ocsw14.ocsw14.local (Interop)	
, ji			

Figure 4-4: Downloaded Topology

4. Expand the Site; right-click on the IP/PSTN Gateway and choose 'New IP/PSTN Gateway'.

🌆 Lync Server 2010 (RC), Topology Builder		
File Action View Help		
🗢 🔿 🖄 🖬 🔽 🗊		
Lync Server 2010 (RC)	The properties for this item are unavailable for editing.	Actions
🖃 🔃 Interop		PSTN gateways
General Edition Front End Servers Enterprise Edition Front End pools		🐁 New IP/PSTN Gateway
Director pools		Topology 🕨
E SQL stores		View ►
		12 Help
Mediation pools		1 Holp
PSIN gatev New IP/PSTN Gateway Monitoring		
Archiving S Topology	•	
Edge pools View	•	
	—	
Top		
🍂 Start 🛛 🏉 🚠 💻 🔢 🔀 Lync Server	2010 (RC	2 10 10 10 10 10 10 10 10 10 10 10 10 10

Figure 4-5: New IP/PSTN Gateway

Figure 4-6: Define New IP/PSTN Gateway

Define New IP/PSTN Gateway		×
Gateway FODN or IP Address *		
E-SBC.OCSW14.local		
Listening port for IP/PSTN gateway: *		
5067		
Sip Transport Protocol: C <u>T</u> CP T I S		
Help	OK Cancel	

 Enter the FQDN of the E-SBC device (i.e. 'E-SBC.OCSW14.local') and click OK. Note that the listening port for the Gateway is '5067' and the transport type is 'TLS'. In certification testing for the AT&T IP Flexible Reach SIP trunk, listening port 5060 was used with transport protocol 'TCP'.

The E-SBC device is now added as an 'IP/PSTN Gateway'.



👪 Lync Server 2010 (RC), Topology Builder			
<u>File Action View H</u> elp			
🗢 🔿 🗾 🖬 🛛 🖬			
Lync Server 2010 (RC)			Actions
Interop Standard Edition Front End Servers	PSIN Gateway		E-SBC.OCSW14.local
Enterprise Edition Front End pools	Cateway FODN or IP	E-SPC OCSW14 local	Edit Properties
Director pools A/V Conferencing pools	Address:	E-500,005W14,008	Topology +
Guilden and the pools SQL stores	Listening port:	5067	View
File stores Madiation and a	SIP Transport Protocol:	TLS	🗙 Delete
PSTN gateways	Alternate media IP address:	Not configured	Help
 gw01.ocsw14.local gw01.ocsw14.local gsBC.OCSW14.local gsBC.OCSW14.local gsC.OCSW14.local gsC.OCSW14.local gsC.OCSW14.local gsC.OCSW14.local gsSBC.OCSW14.local gsSBC	address: Mediation Server	Not associated	rep

Figure 4-7: IP/PSTN Gateway

4.2 Associating the 'IP/PSTN Gateway' with the Mediation Server

This section describes how to associate the 'IP/PSTN Gateway' (E-SBC device) with the Mediation Server.

> To associate the IP/PSTN Gateway with the Mediation Server:

1. Right-click on the **Mediation Server** to use with the IP/PSTN Gateway (i.e. Mediation2.OCSW14.local) and choose **Edit Properties**.

🔀 Lync Server 2010 (RC), Topology Builder			
File Action View Help			
🗢 🔿 🔰 📰 🔢 🗊			
🛃 Lync Server 2010 (RC) ⊡ 🖞 Interop	General	•	Actions Mediation2.ocsw14.local
	FQDN: Associations Edge pool (for media): Note: To view the federation Next hop selection	Mediation2.ocsw14.local <i>Not associated</i> n route, use the site property page.	New Server Edit Properties Topology View X Delete I Help
E-SB Topology Form Form	Next hop pool: Mediation Server PSTN gat	fe-ocsw14.ocsw14.local (Interop)	
Branch since The properties for this pool.	TLS listening port: TCP listening port: PSTN Gateways:	5067 Not configured Default Gateway Site	
🎝 Start 🛛 🏉 🏭 💻 🛛 🙀 Lync Server 2	2010 (RC]	

Figure 4-8: Associating Mediation Server with IP/PSTN Gateway



The following screen is displayed:

Edit Properties		_ 🗆 ×
General	Note: To view or change the federation route, use the site property page.	^
Next hop		
PSTN gateway	Next has selection	
	Next hop pool:	
	fe-ocsw14.local Interop	
	Mediation Server PSTN gateway	•
	Listening ports: * TLS: 5067 TCP:	
	Enable TCP port	
	The following gateways are not associated with any Mediation Server. Click Add to associate them with this	
	Mediation Server.	
	Gateway	
	Med-gw.ocsw14.local Interop	
	E-SBC.OCSW14.local Interop	
	The following gateways are associated with this mediation server. Click New to define a new gateway and add it	
	to the list. Click Remove to remove a gateway from the list.	
	Gateway Site	
	New	
	Deserve	
	Remove	
	Make Default	
		•
Help	ОК	Cancel

Figure 4-9: Before Associating IP/PSTN Gateway to a Mediation Server

2. In the top-left corner, choose **PSTN gateway** and in the Mediation Server PSTN gateway pane, mark the E-SBC device that is designated as the IP/PSTN gateway (i.e. 'E-SBC.OCSW14.local') and click **Add** to associate it with this Mediation Server.

Note that there are two sub-panes, one including a list of IP/PSTN gateways not associated with the Mediation Server and one including a list of IP/PSTN gateways associated with the Mediation server.

Edit Properties	
- cut roperties	
General	Note: To view or change the federation route, use the site property page.
Next hop	
PSTN gateway	Next hop selection
	Next hop pool:
	fe-ocsw14.ocsw14.local Interop
	Mediation Server PSTN gateway
	Listening ports: * TLS: 5067 TCP:
	The following gateways are not associated with any Mediation Server. Click Add to associate them with this Mediation Server.
	Gateway Site
	Med-gw.ocsw14.local Interop Add
	The following gateways are associated with this mediation server. Click New to define a new gateway and add it
	to the list. Click Remove to remove a gateway from the list.
	Gateway Site
	E-SBC.OCSW14.local Interop New
	Remove
	Make Default
Help	OK Cancel

Figure 4-10: After Associating IP/PSTN Gateway to Mediation Server

In the Mediation Server PSTN gateway pane, the IP/PSTN Gateway that you associated with the Mediation Server is displayed with an adjacent Green \checkmark .

3. Click OK.



Lync Server 2010 (RC), Topology Builder				_ 🗆 ×
Eile Action View Help				
🗢 🔿 🗾 🖬 🔢 🖬				
Lync Server 2010 (RC)	General		•	Actions Mediation2.ocsw14.local
B Standard Edition Front End Servers Enterprise Edition Front End pools Director pools SQL stores SQL stores SQL stores Mediation pools E fieldstone foots E fieldstone f	FQDN: Associations Edge pool (for media): Note: To view the federation Next hop selection	Mediation2.ocsw14.local Not associated in route, use the site property page.	•	New Server Edit Properties Topology + View + X Delete Help
E-SBC/OCSVI 4-local E-SBC/OCSVI 4-local E-SBC/OCSVI 4-local B Monitoring Servers G Edde pools E Edde pools E Trusted application servers E Branch sites	Next hop pool: Mediation Server PSTN gat	fe-ocsw14.ocsw14.local (Interop) eway 5067	_	
	TCP listening port: PSTN Gateways:	Not configured Default Gateway Site ✓ E-SBC.OCSW14.local Interop		

Figure 4-11: Media Server PSTN Gateway Association Properties

4. In the Lync Server main menu, choose **Action > Publish Topology**.

لالم المراجع ال	
File Action View Help	
New Central Site	
	Actions
New Topology	Lync Server 2010 (RC) 🔺
Download Topology Default SIP domain: Ocsw14.local	🔢 New Central Site
Bave a copy of Topology As Additional supported SIP Not configured	Edit Properties
Publish Topology domains:	New Topology
Install Database Merae 2007 or 2007 R2 Topology	Open Topology
Remove Deployment Simple URLs	Download Topology
Help	Save a copy of Topology
E B gw01.ocsw14.local Phone access URLs: Active Simple URL	Publish Topology
Med-gw.ocsw14.local Mcd-gw.ocsw14.local SE-SBC.OCSW14.local	Install Database
Monitoring Servers Meeting URLs: <u>Active</u> Simple URL SIP domain	Merge 2007 or 2007 R2 T
Archiving Servers V https://meet.Ocsw14.local Ocsw14.local Ocsw14.local	Remove Deployment
Administrative access Not computed	View 🕨
🗄 🔄 oranch sices	<table-cell> Help</table-cell>
Central Management Server	
Central Management fe-ocsw14.occw14.local (Interop) Server:	
ublish topology to the Central Management Store.	
🕻 Start 🛛 🏀 🚋 🔤 🛛 🙀 Lync Server 2010 (RC	🦉 🇓 🗟 🕼

Figure 4-12: Publishing Topology

The Publish Topology screen is displayed.



Publish Topology	х
Publish the topology	
 In order for Lync Server 2010 (RC) to correctly route messages in your deployment, you must publish your topology. Before you publish the topology, ensure that the following tasks have been completed: A validation check on the root node did not return any errors. A file share has been created for all file stores that you have configured in this topology. All simple URLs have been defined. For Enterprise Edition Front End pools and for Monitoring Servers and Archiving Servers: All SQL stores are installed and accessible remotely; firewall exceptions for remote access to SQL Server are configured. For a single Standard Edition server: The task "Prepare first Standard Edition server" was run. You are currently logged on as a SQL administrator, for example, as a member of the SQL sysadmin role. If you are removing a Front End pool, all users, common area phones, analog devices, application contact objects, and conference directories have been removed from the pool. 	
Help Back Next Cancel	

5. Click Next.

The Topology Builder attempts to publish your topology.

Figure 4-14: Publish Topology Confirmation screen

Pu	blish Topology	(
	Publishing in progress	
	Please wait while Topology Builder tries to publish your topology.	
	Publishing topology	
	Succeeded	
	Downloading topology	
	Succeeded	
	Downloading global simple URL settings.	
	Succeeded	
	Enabling topology	
	Red Net Const	
	<u>B</u> ack <u>N</u> ext Cancel	

Wait until the publish topology process has ended successfully.

Figure 4-15: Publish Topology Successfully Completed

Publish Topology				×
Publishing wizard complete				
Your topology was successfully published.				
Step	Status			
 Publishing topology 	Success			View Logs
Downloading topology	Success			
 Downloading global simple URL settings. 	Success			
 Enabling topology 	Success			
To close the wizard, click Finish.				
Help		Back	<u>F</u> inish	Cancel

6. Click Finish.

4.3 Configuring the 'Route' on the Lync Server 2010

This section describes how to configure a 'Route' on the Lync server and associates it with the IP/PSTN gateway.

To configure the 'route' on the Lync server:

1. Open the Communication Server Control Panel (CSCP), click **Start**, select **All Programs**, and select **Lync Server Control Panel**.



Figure 4-16: Lync Server Control Panel

2. You are prompted for credentials; enter your domain username and password.



Figure 4-17: Lync Server Credentials

The CSCP Home page is displayed.

ì	Home		
	Users		
	Topology	User Information	Resources
	IM and Presence	Welcome, Administrator	Getting Started
	Voice Routing	View your roles	Using Control Panel
	Voice Features	Ton Actions	Microsoft Lync Server 2010
	Response Groups		Getting Help Downloadable Documentation
	Conferencing	Enable users for Lync Server Edit or move users	Online Documentation on TechNet Library
	Clients	View topology status	Lync Server Management Shell Script Library
	External User	View Monitoring Server reports	
	Access		Forums
	Monitoring		Blogs
	Security		
	Network		
	Configuration		

Figure 4-18: CSCP Home page

3. In the Navigation pane, select the 'Voice Routing' option.

🌄 Mie	crosoft Lync Server 2010	Control Pan	el						
8	Microsoft*								Administrator Sign out
Ø.	Lync Server 20	10							4.0.7457.0
	Home	Dial Pl	an Voice Policy	Route P	STN Usage	Trunk Conf	iguration Test Voice R	louting	
33	Users	Crea	te voice routing te	st case inforn	nation				*
×	Topology								
₽	IM and Presence						Q		
ণ্ড	Voice Routing	🗣 N	ew 🔻 🧪 Edit 🔻	Action •	Commit	•			0
S	Voice Features		Name		Scope	State	Normalization rules	Description	
23	Response Groups		決 Global		Global	Committed	1		
Ð	Conferencing		LAB2		Site	Committed	1		
	Clients		SBA-LAB3		Site	Committed	2		
B	External User Access	•							
	Monitoring and Archiving								
9	Security								
9	Network Configuration								

Figure 4-19: Voice Routing Option

4. In the Voice Routing menu at the top of the page, select the **Route** option.



	Server 2010 Control P	Panel				
Microsoft*						Administ
Lync's	erver2010					
Home	Dia	al Plan Voice Policy	Route PSTN Usage Tru	nk Configuration	Test Voice Routing	
Users	C	Create voice routing te	st case information			
Topology						
IM and Pre	esence				Q	
Voice Rou	ting	New 🥖 Edit 🔻	The Move up 🔒 Move down	Action V Comm	nit 🔻	
, Voice Feat	tures	Name	State	PSTN usage	Pattern to match	
Response	Groups	LocalRoute	Committed	Internal, Local	^((\+1[0-9]{10}) (\+972) (\+011))	
Conference	ing					
Clients						
External U	ser					
Access						
Monitorin and Archiv	g ving					
Security						
Network						
Continues						

Figure 4-20: Route Option

- 5. In the content area toolbar, click
- 6. In the Build a Pattern to Match pane, fill in a Name for this route (i.e SIP Trunk Route) and a Pattern to Match for the phone numbers you wish this route to handle. In this example, the pattern to match is '*', which implies "to match all numbers".
- 7. Click Add.

licrosoft Lync Server 2010 Co	trol Panel	
Microsoft*		Administrator Sign
Lync Server 2010		4.0.74
Home	Dial Plan Voice Policy Route PSTN Usage Trunk Configuration Test Voice Routing	
Users	Create voice routing test case information	♦
Topology		
IM and Presence	New Voice Route	
Voice Routing	√ OK X Cancel	0
Voice Features	Name:*	-
Response Groups	SIP Trunk Route	
Conferencing	Description:	
Clients		
External User	Build a Pattern to Match Add the starting digits that you want this route to handle, or create	
Access	the expression manually by clicking Edit.	
Monitoring	Starting digits for numbers that you want to allow:	
and Archiving		
Security	Exceptions	
Network	1 State State	
consignation		
	Match this pattern:*	
	Edit Reset	

Figure 4-21: Adding New Voice Route

8. Associate the route with the IP/PSTN gateway you created above; scroll down to the Associated Gateways pane and click **Add**.

A list of all the deployed Gateways is displayed.

		Q
Service	Site	
stnGateway:gw01.ocsw14.local	Interop	
PstnGateway:SBA-gw.OCSW14.local	M1K	
ostnGateway:SBA-gw2.OCSW14.local	M2K	
ostnGateway:sba-gw03.ocsw14.local	M2K-Test	
ostnGateway:ofer-gw.ocsw14.local	ofer	
stnGateway:GW-LAB2.ocsw14.iocal	LAB2	
stnGateway:GW-LAB3.ocsw14.local	SBA-LAB3	
PstnGateway:Med-gw.ocsw14.local	Interop	
PstnGateway: E-SBC.OCSW14.local	Interop	

Figure 4-22: List of Deployed Gateways

9. Select the IP/PSTN Gateway you created above and click OK.

Figure 4-23: Selecting the PSTN Gateway

Nie Mie	rosoft Lync Server 2010	Control Panel	
	Microsoft		Administrator Sign out
Ø.	Lync Server 20	10	4.0.7457.0
	Home	Dial Plan Voice Policy Route PSTN Usage Trunk Configuration Test Voice Routing	
33	Users	Create voice routing test case information	*
×	Topology		
Ģ	IM and Presence	New Voice Route	
હ	Voice Routing	✓ OK X Cancel	
C	Voice Features	Edit Reset 🕐	-
23	Response Groups		
Ð	Conferencing	Suppress caller ID	
6	Clients	Alternate carer ID:	
4	External User	Associated gateways:	
	Access	PstnGateway:E-SBC.OCSW14.local Add	
	and Archiving	Remove	
•	Security		
9	Network	Associated PSTN Usages	
	Conliguration	Select Remove 1	
		PSTN usage record Associated voice policies	
			_

10. Associate a PSTN Usage to this route. In the **Associated PSTN Usages** toolbar, click **Select** and add the associated PSTN Usage.

Microsoft Lync Server 2010 Co	ntrol Panel	_
Microsoft"		Administrator Sig
Lync Server 2010		4.0.7
Home	Dial Plan Voice Policy Route PSTN Usage Trunk Configuration Test Voice Routing	
Users	Create voice routing test case information	8
Topology		
IM and Presence	New Voice Route	
Voice Routing	√ OK X Cancel	0
V: F .	Associated gateways:	^
voice Features	PstnGateway:E-SBC.OCSW14.local Add	
Response Groups	Remove	
Conferencing		
Clients		
External User	Associated PSTN Usages	
Access	Select Remove 🏠 🥾	
Monitoring	PSTN usage record Associated voice policies	
and Archiving	Internal 💮 Global	
Security	Local 💮 Global	
Network Configuration	Long Distance 💮 Global	
	Translated number to test:	
	Go	
		-

Figure 4-24: Associating PSTN Usage to PSTN Gateway

11. Click the **OK** button in the toolbar at the top of the New Voice Route pane.

Wicrosoft Lync Server 2010 Control Panel								
Ø.	Lync Server 201	0						Administrator Sign out 4.0.7457.0
	Home	Dial Plan	Voice Policy	Route PSTN Usag	ge Truni	k Configuration	Test Voice Routing	
33	Users	Create	voice routing te	st case information				*
м	Topology							
₽	IM and Presence						<u>P</u>	
ণ্ড	Voice Routing	🗣 New	🧪 Edit 🔻	🏠 Move up 🛛 🕹 Mov	e down	Action 🔻 Com	mit 🔻	0
6	Voice Features	Nan	ne	State		PSTN usage	Pattern to match	
23	Response Groups	Loca	IRoute	Commit	tted	Internal, Local	^((\+1[0-9]{10}))(\+972))(\+011))	
₽	Conferencing	SIP 1	Frunk Route	1 Unc	ommitted	Internal, Local	* •	
	Clients							
14	External User Access							
-	Monitoring and Archiving							
9	Security							
Ŷ	Network Configuration							
1					_			

Figure 4-25: Confirmation of New Voice Route

12. In the Content area Toolbar, click on the arrow adjacent to the **Commit** button; a drop-down menu is displayed; select the 'Commit All' option.

Figure 4-26: Committing Voice Routes

						trol Panel	10 Conl	crosoft Lync Server 2010	🌄 Mi
nistrator Sign out	Admir						2010	Lync Server 20	2
4.0.7457.		/oice Routing	ation	unk Configura	PSTN Usage Tru	Dial Plan Voice Policy Route			100
*					ormation	Create voice routing test case info		Home	22
						-	-	Topology	89 54
								IM and Presence	Ģ
			Comm	Action X	n A Move down	🗗 Nau 🧷 Edit 🔻 🔶 Maya u		Voice Routing	(e
v		committed changes	sa Revie	PSTN us	State	Name		Voice Features	C
)	11))	L Comr	Internal,	Committed	LocalRoute		Response Groups	23
		cted changes	L Cance	i Internal,	1 Uncommitted	SIP Trunk Route		Conferencing	Ŗ
		ncommitted changes	Cance					Clients	e
								External User Access	論
								Monitoring and Archiving	
								Security	8
								Network Configuration	9

13. In the Uncommitted Voice Configuration Settings window, click **Commit**.

guration Setting:	\$		
			*
Action	New value (pattern to match)	Old value (pattern to match)	
Modified	*	1	
	guration Setting: Action Modified	guration Settings Action New value (pattern to match) Modified *	Action New value (pattern to match) Old value (pattern to match) Modified * *

Figure 4-27: Uncommitted Voice Configuration Settings



14. A message is displayed, confirming a successful voice routing configuration; in the Microsoft Lync Server 2010 Control Panel prompt, click Close.



Figure 4-28: Voice Routing Configuration Confirmation

The new committed Route is now displayed in the Voice Routing screen.

Figure 4-29: Voice Routing Screen Displaying Committed Routes

Nie Mie	crosoft Lync Server 2010 Co	ntrol Panel				
ð.	Lync ⁻ Server2010					Administrator Sign out 4.0.7457.0
	Home	Dial Plan Voice Policy	Route PSTN Usage Tru	ink Configuration	Test Voice Routing	
33	Users	Create voice routing test	case information			*
×	Topology					
₽	IM and Presence				<u>P</u>	
ণ্ড	Voice Routing	🗣 New 🧪 Edit 🔻 👚	Move up 🕹 Move down	Action ▼ Com	mit 🔻	0
6	Voice Features	Name	State	PSTN usage	Pattern to match	
23	Response Groups	LocalRoute	Committed	Internal, Local	^((\+1[0-9]{10}) (\+972) (\+011))	
Ð	Conferencing	SIP Trunk Route	Committed	Internal, Local	*	
6	Clients					
ł	External User Access					
	Monitoring and Archiving					
9	Security					
9	Network Configuration					
J						
5 **Configuring E-SBC Device**

This section provides step-by-step procedures for configuring the E-SBC device. The following describes the steps required to configure the E-SBC device :

- **Step 1**: Configure IP Addresses. See section 5.1 on page 38.
- **Step 2**: Enable the SBC Capabilities. See section 5.2 on page 39.
- **Step 3**: Configure the Number of Media Channels. See section 5.3 on page 40.
- **Step 4**: Configure the Proxy Sets. See section 5.4 on page 41.
- **Step 5**: Configure the IP Groups. See section 5.5 on page 43.
- **Step 6**: Configure the Voice Coders. See section 5.6 on page 45.
- Step 7: Define Silence Suppression and Comfort Noise. See section 5.6.1 on page 46.
- **Step 8**: Configure IP Profile Settings. See section 5.7 on page 47.
- **Step 9**: Configure IP-to-IP Routing Setup. See section 5.8 on page 49.
- **Step 10**: Configure Number Manipulation. See section 5.9 on page 52.
- **Step 11**: Configuring IP Profile for Call Forwarding. See section 5.10 on page 57.
- **Step 12**: Configuring SIP General Parameters. See section 5.11 on page 60.
- **Step 13**: Defining Reasons for Alternative Routing. See section 5.12 on page 62.

The procedures described in this section are performed using the E-SBC devices' Web-based management tool (i.e., embedded Web server). Before you begin configuring the E-SBC device, ensure that the Web interface's Navigation tree is in full menu display mode (i.e., the **Full** option on the Navigation bar is selected), as displayed below:

Configuration Management Status & Diagnostics Configuration Management Status & Diagnostics Scenarios Search Scenarios Search 🗩 Basic 🔘 Full \bigcirc 🔾 Basic 🧿 Full Basic Full Navigation 🗉 🧰 Network Settings Navigation Tree ⊕ i Media Settings Tree View View Option Option 🗉 🧰 Protocol Configuration E Advance Applications Protocol Configuration Advance Applications Only "Basic" Menus All Menus

Figure 5-1: Web Interface Showing Basic/Full Navigation Tree Display

5.1 Step 1: Configure IP Addresses

This section describes how to configure IP addresses when a single LAN interface is used to connect to the Flexible Reach SIP Trunk. In this configuration, the internal data-routing capabilities of the E-SBC device are not used. As a consequence, you must disable the internal data-routing interface as described in the procedure below.



Note: When operating in LAN VoIP-only mode, do not use the E-SBC device's WAN port.

\blacktriangleright To operate the E-SBC device as a LAN VoIP gateway only:

- 1. Disconnect the network cable from the WAN port and then connect one of the E-SBC device's LAN ports to the network.
- 2. Disable or remove the data-routing IP network interface:
 - Access the 'Connections' page (Configuration tab > Data menu > Data System > Connections).
 - Delete the "LAN Switch VLAN 1" connection by clicking the corresponding Remove button, and then clicking OK to confirm deletion.

Figure 5-2: Removing Data-Routing Connection Interface



- Configure VoIP IP network interfaces in the 'Multiple Interface' table (Configuration tab > VoIP menu > Network > IP Settings).
 - In the 'Multiple Interface' table, define a single IP network interface for application types "OAMP + Media + Control".

Figure 5-3: Multiple Interface Table

Index	Application Type	IP Address	Prefix	Gateway	VLAN ID	Interface Name	Primary DNS Server IP Address	Secondary DN IP Addre
0 💌	OAMP + Media + Control 👻	10.15.9.118	16	10.15.0.0	1	Voice	0.0.0	0.0.0

• Click **OK** to save settings.

5.2 Step 2: Enable the SIP SBC Application Mode

This step describes how to enable the gateway-SBC devices' SIP SBC application mode.

To enable the SIP SBC application mode:

 Open the 'Application Enabling' page (Configuration tab > VoIP menu > Applications Enabling > Applications Enabling).

Figure 5-4: Application Enabling

•		
🗲 Enable SAS	Disable	•
Enable SBC Application	Disable	· 🖉
🗲 Enable IP2IP Application	Enable	· <mark>- 2</mark>

 From the 'Enable IP2IP Application' drop-down list, select "Enable". Reset with BURN to FLASH is required.



Note: To enable the IP2IP capabilities on the AudioCodes device, your device must be loaded with the feature key that includes the IP2IP feature and also the E-SBC device must be running SIP version 6.2 or later.

5.3 Step 3: Configure the Number of Media Channels

In order to reform the coder transcoding, you need to define DSP channels. The number of media channels represents the number of digital signaling processors (DSP) channels that the device allocates to IP-to-IP calls (the remaining DSP channels can be used for PSTN calls). Two IP media channels are used per IP-to-IP call. The maximum number of media channels available on the Mediant 1000 E-SBC device is 120 (i.e., up to 60 IP-to-IP calls). The maximum number of media channels available on the Mediant 3000 E-SBC Media Gateway device is 2016 (i.e., up to 1008 IP-to-IP calls).

To configure the number of media channels:

 Open the 'IP Media Settings' page (Configuration tab > VoIP menu > IP Media > IP Media Settings).

					Basic Parame
•					
4	Number of Media Channels	2 →	120		2
4	Voice Streaming	Ŭ	Disable	~	
	NetAnn Announcement ID		annc		
	MSCML ID		ivr		
	Transcoding ID		trans		
•	Conference				
	Conference ID		conf		
	Beep on Conference		Enable	*	
	Enable Conference DTMF Clamping		Enable	*	
	Enable Conference DTMF Reporting		Disable	*	

Figure 5-5: IP Media Channels Settings

2. In the 'Number of Media Channels', enter "120" to enable up to 60 IP-to-IP calls with transcoding. Click **Apply New Value**.

5.4 Step 4: Configure the Proxy Sets

This step describes how to configure the Proxy Sets. The Proxy Sets represent the IP addresses (or FQDN), which are required for communicating with the entities in the network:

- Proxy Set ID #1 is assigned with the IP address of AT&T IP Flexible Reach SIP Trunk.
- Proxy Set ID #2 is assigned with the IP address of Lync Mediation server.

These Proxy Sets are later assigned to IP Groups (see Section 5.5 on page 43).

To configure proxy sets:

- Open the 'Proxy Sets Table' page (Configuration tab > VoIP menu > Control Network > Proxy Sets Table).
- 2. Configure the Proxy Set for AT&T IP Flexible Reach SIP Trunk:

From the 'Proxy Set ID' drop-down list, select "1".

- a. In the 'Proxy Address' column, enter the IP address or FQDN of the AT&T IP Flexible Reach SIP Trunk and the listening port of the AT&T IP Flexible Reach SIP Trunk.
- **b.** From the 'Transport Type' drop-down list, corresponding to the IP address entered above, select "UDP".
- **c.** Repeat steps 'a' and 'b' as required for alternate AT&T IP Border Element (if used).

Figure 5-6: Proxy Set ID 1 for AT&T IP Flexible Reach SIP Trunk

roxy Set	ID	1	×
	Proxy Address		Transport Type
1	207.242.225.210:5060		
2			~
3			~
4			~
5			✓

AudioCodes

- **3.** Configure the Proxy Set for the Lync Mediation Server:
 - a. From the 'Proxy Set ID' drop-down list, select "2".
 - **b.** In the 'Proxy Address' column, enter the IP address or the FQDN and the listening port of the Lync Mediation Server.
 - **c.** From the 'Transport Type' drop-down list corresponding to the IP address entered above, select "TCP" Transport Type.

Figure 5-7: Proxy Set ID 2 for Lync Mediation Server

Proxy S	et II)	2	~
		Proxy Address		Transport Type
	1	10.15.9.11:5060		
	2			
	3			·
	4			
	5			

5.5 Step 5: Configure the IP Groups

This step describes how to create IP groups. Each IP group represents a SIP entity in the device's network. You need to create IP groups for the following entities:

- 1. AT&T IP Flexible Reach SIP trunk
- 2. Lync Server 2010 Mediation Server

These IP groups are later used by the IP2IP application for routing calls.

> To configure IP Groups:

 Open the 'IP Group Table' page (Configuration tab > VoIP menu > Control Network> IP Group Table).

•	-		
	Index 2a	1	✓
-	Common Parameters		
	Туре	SERVER	 ✓ ✓
	Description 20	ATT	
	Proxy Set ID	1	2d
	SIP Group Name 2e	207.242.225.210	
	Contact User		
4	SRD	0	
4	Media Realm		~
	IP Profile ID (2f)	1	~
	Colomba Demonstration		
•	Gateway Parameters		
	Always Use Route Table	No	*
	Routing Mode	Not Configured	*
	SIP Re-Routing Mode	Standard	~
	Enable Survivability	Disable	~
	a 1 ma m		

Figure 5-8: IP Group 1 Table

- 2. Define IP Group #1 for the AT&T IP Flexible Reach SIP Trunk as follows:
 - a. IP Group Index '1'
 - b. Type: "SERVER"
 - c. Description: arbitrary name. (e.g., "ATT")
 - **d.** Proxy Set ID: "1" (represents the IP address, configured in Section 5.4 on page 41, for communicating with this IP Group).
 - e. SIP Group Name: The SIP Request-URI host name used in INVITE messages sent to the IP Group, or the host name in the From header of INVITE messages received from the IP Group. Enter the WAN IP address.
 - f. IP Profile ID: "1": Different IP profile is used for the AT&T IP Flexible Reach SIP Trunk and the Mediation Server. See Section 5.7 on page 47.



			Basic Paramete
•		<u>_</u>	~
	Index	(3a) 2 V	
		Ŭ	
•	Common Parameters		
	Туре	server de server	
	Description		
	Proxy Set ID	2 3d	=
	SIP Group Name	(3e) 100.33.2.105	
	Contact User		
\$	SRD	0	
4	Media Realm	×	
	IP Profile ID	(3f) → 2	
Ŧ	Gateway Parameters		
	Always Use Route Table	No	
	Routing Mode	Not Configured	
	SIP Re-Routing Mode	Standard 🗸	
	Enable Survivability	Disable	
	a 1 10 a 10		

Figure 5-9: IP Group 2 Table Page

- **3.** Define IP Group **#2** for Mediation Server as follows:
 - a. Select IP Group Index '2':
 - b. Type: "SERVER"
 - c. Description: <Free Description> (e.g., "Lync Mediation Server")
 - d. Proxy Set ID: "2"
 - e. SIP Group Name: The SIP Request-URI host name used in INVITE messages sent to the IP Group, or the host name in the From header of INVITE messages received from the IP Group. Enter the Gateway Name.
 - f. IP Profile ID: "2" (see Section 5.7 on page 47).

5.6 Step 6: Configure the Voice Coders

Since the Mediation Server supports both the G.711A-law and G.711U-law voice coders, while the AT&T Flexible Reach SIP trunk requires the G.711U-law coder, you can configure a single coder table reference for both services by utilizing the G711U-law coder. The Coder table is associated with an IP Profile. Both IP Profile indices 1 & 2 referenced in this document, reference 'Default Coder Table' (see Section 5.7 on page 47) which is associated with the IP Group (see Section 5.5 on page 43).

To configure Coder Table for Mediation server and AT&T Flexible Reach SIP Trunk:

 Open the 'Coders Table' page (Configuration tab > VoIP menu > Coders And Profiles > Coders).

Coder Name	Packetization Time	Rate	Payload Type	Silence Suppression
G.711U-law (3)	20 💌	64 💙	0	Disabled 4
~	*	~		×
~	~	~		~
~	~	~		~
×	~	~		~
~	~	~		~
~	~	~		~
~	~	~		~
~	~	~		~
×	~	~		×

Figure 5-10: Coder Group Table - Mediation Server

- 2. From the 'Coders Table' prepare to select via drop-down list, coder and attributes.
- 3. Select the G.711U-law coder, as shown in the figure above.
- **4.** From 'Silence Suppression' drop-down list, select 'Enable' or 'Disabled' as shown in the figure above.

5.6.1 Step 7: Define Silence Suppression and Comfort Noise

Overall voice quality has been significantly improved for the Microsoft Lync 2010 environment. These improvements include suppression of typing noise during calls and improved generation of "comfort noise," which reduces hissing and smoothes over the discontinuous flow of audio packets. You may need to modify the Silence Suppression and Comfort Noise parameters to achieve this goal. Note that the Echo canceller is enabled by default.

To configure silence suppression parameters:

- 1. Silence Suppression is configured per coder type. (See Section 5.6 on page 45 above to enable Silence Suppression per coder.)
- Open the 'RTP/RTCP Settings' page (Configuration tab > Media menu > RTP / RTCP Settings).

•	General Settings	
	Dynamic Jitter Buffer Minimum Delay	10
	Dynamic Jitter Buffer Optimization Factor	10
	RTP Redundancy Depth	0
	Packing Factor	1
	Basic RTP Packet Interval	Default 🗸
	RFC 2833 TX Payload Type	101
	RFC 2833 RX Payload Type	101
	RFC 2198 Payload Type	104
	Fax Bypass Payload Type	102
	Enable RFC 3389 CN Payload Type	Enable 🗸
	Comfort Noise Generation Negotiation	Enable 3
	Remote RTP Base UDP Port	0
4	RTP Multiplexing Local UDP Port	0
4	RTP Multiplexing Remote UDP Port	0
4	RTP Base UDP Port	16400 4

Figure 5-11: RTP/RTCP Settings Page

- **3.** From the 'Comfort Noise Generation Negotiation' drop-down list, select 'Enable'. This action enables negotiation and usage of Comfort Noise (CN).
- 4. Ensure 'RTP Base UDP Port' is '16400' to ensure that the RTP is within the port range of 16384-32767 (port range required by the Cisco Router with IP Flexible Reach-MIS/PNT/AVPN).
- 5. Click Submit.

5.7 Step 8: Configure IP Profile Settings

This section describes how to configure the IP Profile Settings.

> To configure IP Profile for AT&T IP Flexible Reach:

 Open the 'IP Profile Settings' page (Configuration tab > VoIP menu > Coders and Profiles > IP Profile Settings).

Profile ID ~ ▶ 1 (2)Profile Name Common Parameters Gateway Parameters No Fax Fax Signaling Method V Play Ringback Tone to IP Don't Play ¥ Enable Early Media Enable × Copy Destination Number to Redirect Number Disable v Media Security Behavior Disable ¥ CNG Detector Mode Disable Y Modems Transport Type Enable Bypass × NSE Mode Disable × Number of Calls Limit -1 Progress Indicator to IP Not Configured Y Profile Preference 1 × Coder Group (3) Default Coder Group Y Remote RTP Base UDP Port 0 First Tx DTMF Option RFC 2833 ~ Second Tx DTMF Option Y Declare RFC 2833 in SDP Yes ¥ Add IE In SETUP AMD Sensitivity Parameter Suit 0 8 AMD Sensitivity Level 300 AMD Max Greeting Time AMD Max Post Silence Greeting Time 400 Enable Hold Enable ¥

Figure 5-12: IP Profile Page-AT&T IP Flexible Reach Server

- 2. From the 'Profile ID' drop-down list, select '1'.
- 3. From the 'Coder Group' drop-down list, select 'Default Coder Group'.

To configure IP Profile for Mediation Server:

 Open the 'IP Profile Settings' page (Configuration tab > VoIP menu > Coders and Profiles > IP Profile Settings).

Figure 5-13:	IP Profile	Page-Mediation	Server

Profile ID	(2)	2	*
Profile Name		ocs	
Common Parameters			
Gateway Parameters		N. 5	
Fax Signaling Method		No Fax	~
Play Ringback Tone to IP		Don't Play	*
Enable Early Media		Enable	*
Copy Destination Number to Redirect Number	\sim	Disable	~
Media Security Behavior	3	Disable	*
CNG Detector Mode		Disable	~
Modems Transport Type		Enable Bypass	~
NSE Mode		Disable	*
Number of Calls Limit		-1	
Progress Indicator to IP		PI = 8	~
Profile Preference		1	~
Coder Group	4-	Default Coder Group	~
Remote RTP Base UDP Port		0	
First Tx DTMF Option		RFC 2833	*
Second Tx DTMF Option			*
Declare RFC 2833 in SDP		Yes	*
Add IE In SETUP			
AMD Sensitivity Parameter Suit		0	
AMD Sensitivity Level		8	
AMD Max Greeting Time		300	
AMD Max Post Silence Greeting Time		400	
Enable Hold		Enable	*

- 2. From the 'Profile ID' drop-down list, select '2'.
- **3.** From the 'Media Security Behavior' drop-down list, select one of the following options:
 - "Mandatory" if Mediation Server is configured to SRTP Required
 - "Preferable-Single media" if Mediation Server is configured to SRTP Optional.
 - "Disable" if the Mediation Server is configured to SRTP disabled.
- 4. From the 'Coder Group' drop-down list, select 'Default Coder Group'.

5.8 Step 9: Configure IP-to-IP Routing Setup

The E-SBC devices' IP-to-IP call routing capabilities is performed in two stages:

- 1. Inbound IP Routing: Recognizes the received call as an IP-to-IP call, based on the call's source IP address. This step is configured in the 'Inbound IP Routing Table'
- 2. Outbound IP Routing: Once recognized as an IP-to-IP call in the first stage (see above), the call is routed to the appropriate destination (i.e., IP address). This step is configured in the 'Outbound IP Routing Table'.

5.8.1 Configure Inbound IP Routing

This step defines how to configure the E-SBC device for routing inbound (i.e., received) IP-to-IP calls. The figure shown below illustrates three different call scenarios, corresponding to Index #1, Index #2 and Index#3 (described below).

> To configure inbound IP routing:

 Open the 'Inbound IP Routing Table' page (Configuration tab > VoIP menu > GW and IP to IP > Routing submenu > IP to Trunk Group Routing).

V								
	Routing Index	Routing Index		1-10 💌				
	IP To Tel Routing	IP To Tel Routing Mode		Route calls before manipulation 😒				
					_			
Dest. Host Prefix	Source Host Prefix	Dest. Phone Prefix	Source Phone Prefix	Source IP Address	+	Group ID	IP Profile ID	Source IPGroup ID
1 2		*	+1214291	10.15.9.11		-1	2	2
2 - 3	FE-Lync.Lync.local	*	*	10.15.9.11		-1	3	2
3-4-4		*	*	207.242.225.210		-1	1	1

Figure 5-14: Inbound IP Routing Table Page

- Index #1 configuration identifies all IP calls received from the Mediation Server as IP-to-IP calls and assigns them to the IP Group ID configured for the Lync Mediation Server as verified Lync assigned telephone numbers within a prefix range:
 - 'Dest Phone Prefix': Enter the asterisk (*) symbol to indicate all destinations.
 - 'Source Phone Prefix': Enter the Lync assigned telephone prefix to screen for vaid direct IP-toIP calls.
 - 'Source IP Address': Enter the IP address of the Mediation server.
 - 'Trunk Group ID': Enter "-1" to indicate that these calls are IP-to-IP calls.
 - 'IP Profile ID: Enter '2' indicate the IP Profile for Mediation server.
 - 'Source IP Group ID': Enter "2" to assign these calls to the IP Group pertaining to the Mediation server.
- 3. Index #2 configuration identifies all IP calls received from the Mediation Server in the event of a call forwarding Scenario (see section 5.10 on page 57) as IP-to-IP calls and assigns them to the IP Group ID configured for the Mediation server:
 - 'Source Host Prefix: Enter the Lync Front end FQDN in case of call forwarding, the Source host in the incoming INVITE from the Mediation Server is the Lync Front End server FQDN, while for regular calls, the Source host is the Mediation Server FQDN.
 - 'Dest Phone Prefix': Enter the asterisk (*) symbol to indicate all destinations.
 - 'Source IP Address': Enter the IP address of Mediation Server.
 - 'Trunk Group ID': Enter "-1" to indicate that these calls are IP-to-IP calls.
 - 'IP Profile ID: Enter '3' to indicate that the IP Profile supports the call forwarding scenario.
 - 'Source IP Group ID': Enter "2" to assign these calls to the IP Group pertaining to the Mediation server.
- 4. Index #3 configuration identifies all IP calls received from AT&T IP Flexible Reach SIP Trunk as IP-to-IP calls and assigns them to the IP Group ID configured for the AT&T IP Flexible Reach SIP Trunk:
 - 'Dest Phone Prefix': Enter the asterisk (*) symbol to indicate all destinations.
 - 'Source IP Address': Enter the IP address of AT&T IP Flexible Reach SIP Trunk.
 - 'Trunk Group ID': Enter "-1" to indicate that these calls are IP-to-IP calls.
 - 'IP Profile ID: Enter '1' indicate the IP Profile for AT&T IP Flexible Reach SIP Trunk.
 - 'Source IP Group ID': Enter "1" to assign these calls to the IP Group pertaining to AT&T IP Flexible Reach SIP Trunk.

5.8.2 Configure Outbound IP Routing

This step defines how to configure the E-SBC device for outbound routing (i.e., sent) IP-to-IP calls. The figure shown below illustrates two different call scenarios, corresponding to Index #1 and Index #2 (described below).

To configure outbound IP routing:

 Open the 'Outbound IP Routing Table' page (Configuration tab > VoIP menu > GW and IP to IP > Routing submenu > Tel to IP Routing).

Figure 5-15: Outbound IP Routing Table Page

	•								
	Routing Index				1-10	•			
	Tel To IP Routin	ng Mode			Route cal	ls before manipulation 🔻			
	L								
Src. IPGroupID Src. Host Prefix De	est Host Prefix	Src. Trunk Group ID	Dest. Phone Prefix	Source Phone Pre	efix >	Dest. IP Address	Port	Transport Type	Dest. IPGroup
1 1 2	*	•	*	*				Not Configured 💌	2
2 2 -3	*	•	*	*				Not Configured 👻	1

- Index #1 defines routing of IP calls to the Lync 2010 Mediation server. All calls received from Source IP Group ID 1 (i.e., from the AT&T IP Flexible Reach SIP trunk) are routed to Destination IP Group ID 2 (i.e., to Lync 2010 Mediation server):
 - 'Source IP Group ID': Select "1" to indicate received (inbound) calls identified as belonging to the IP Group configured for the AT&T IP Flexible Reach SIP trunk.
 - 'Dest Phone Prefix': Enter the asterisk (*) symbol to indicate all destinations.
 - 'Source Phone Prefix': Enter the asterisk (*) symbol to indicate all callers.
 - 'Dest IP Group ID': Select "2" to indicate the destination IP Group to where the calls must be sent, i.e., to Lync 2010 Mediation server.
- 3. Index #2 defines the routing of IP calls to the AT&T IP Flexible Reach SIP Trunk. All calls received from IP Group ID 2 (i.e., Lync 2010 Mediation server) are routed to Destination IP Group ID 1 (i.e., AT&T IP Flexible Reach SIP Trunk):
 - 'Source IP Group ID': Select "2" to indicate received (inbound) calls identified as belonging to the IP Group configured for the Lync 2010 Mediation Server 'Dest Phone Prefix': Enter the asterisk (*) symbol to indicate all destinations.
 - 'Source Phone Prefix': Enter the asterisk (*) symbol to indicate all callers.
 - 'Dest IP Group ID': Select "1" to indicate the destination IP Group to where the calls must be sent, i.e., to the AT&T IP Flexible Reach SIP Trunk.

5.9 Step 10: Configure Number Manipulation

The Manipulation Tables submenu allows you to configure number manipulation and mapping of NPI/TON to SIP messages. This submenu includes the following options:

- Dest Number IP->Tel. See Section 5.9.1 on page 53.
- Dest Number Tel->IP. See Section 5.9.1 on page 53.
- Source Number IP->Tel. See Section 5.9.2 on page 55.
- Source Number Tel->IP. See Section 5.9.2 on page 55.

5.9.1 Configure Destination Phone Number Manipulation

This section describes how to configure the destination phone number manipulation.

To configure Destination Phone Number Manipulation Table for IP -> Tel Calls Table:

 Open the 'Destination Phone Number Manipulation Table for IP -> Tel calls' page (Configuration tab > VoIP menu > GW and IP to IP > Manipulations submenu > Dest Number IP >Tel).

Figure 5-16: Destination Phone Number Manipulation Table for IP -> Tel Calls Page

Index	Destination Prefix	Source Prefix	Source IP Address	Stripped Digits From Left	Stripped Digits From Right	Prefix to Add	Suffix to Add	Num
1 ()	+1	•	10.15.9.11	2	0	1		255
2 ()	+0	•	10.15.9.11	1	0			255
з О	+	•	10.15.9.11	1	0	1		255
4 ()	1	*	10.15.9.11	1	0	1		255

- Index #1 defines destination number manipulation of IP calls from Lync 2010 Mediation server. All calls received from IP address 10.15.9.11 (i.e., from Mediation Server) and the destination number prefix begins with '+1', Remove the '+1' from the Number, and add the prefix '1'.
- Index #2 defines destination number manipulation of IP calls from Lync 2010 Mediation server. All calls received from IP address 10.15.9.11 (i.e., from Mediation Server) and the destination number prefix begins with '+0', Remove the '+' from the Number.
- Index #3 defines destination number manipulation of IP calls from Lync 2010 Mediation server. All calls received from IP address 10.15.9.11 (i.e., from Mediation Server) and the destination number prefix begins with '+', Remove the '+' from the Number, and add the prefix '1'.
- Index #4 defines destination number manipulation of IP calls from Lync 2010 Mediation server. All calls received from IP address 10.15.9.11 (i.e., from Mediation Server) and the destination number prefix begins with '1', Remove the '1' from the Number, and add the prefix '1'.

To configure Destination Phone Number Manipulation Table for Tel -> IP Calls Table:

 Open the 'Destination Phone Number Manipulation Table for Tel -> IP calls' page (Configuration tab > VoIP menu > GW and IP to IP > Manipulations submenu > Dest Number Tel->IP).

In	dex	Source Trunk Group	Source IP Group	Destination Prefix	Source Prefix	Stripped Digits From Left	Stripped Digits From Right	Prefix to Add	Suffix to Add	Number of Digit Leave
0	0	-1	1	+	•	0	0			255
1	0	-1	1	1	•	0	0	+		255
2	0	-1	1	3684891	+	7	0	+12142911002		255
3	0	-1	1	XXXXXXXXXXX#	+	0	0	+1		255
4	0	-1	2	1511	•	1	0			255
5	0	-1	2	1911	•	1	0			255

Figure 5-17: Destination Phone Number Manipulation Table for Tel -> IP Calls Page

- **Index #0** defines destination number manipulation of IP calls from AT&T IP Flexible Reach SIP Trunk. All calls received from Source IP Group 1 (i.e., from AT&T IP Flexible Reach SIP Trunk) and the destination number prefix begins with '+', do not perform any changes to the number.
- Index #1 defines destination number manipulation of IP calls from AT&T IP Flexible Reach SIP Trunk. All calls received from Source IP Group 1 (i.e., from AT&T IP Flexible Reach SIP Trunk) and the destination number prefix begins with '1', add the '+' prefix to the number.
- Index #2 defines destination number manipulation of IP calls from AT&T IP Flexible Reach SIP Trunk. All calls received from Source IP Group 1 (i.e., from AT&T IP Flexible Reach SIP Trunk) and the destination number prefix begins with '3684891' as a 7 digit private number, remove the 7 digits, and add the '+12142911002' as the new 10 digit number.
- Index #3 defines destination number manipulation of IP calls from AT&T IP Flexible Reach SIP Trunk. All calls received from Source IP Group 1 (i.e., from AT&T IP Flexible Reach SIP Trunk) and the destination number length is 10 digit number, add the '+1' prefix to the number.
- Index #4 defines destination number manipulation of IP calls from AT&T IP Flexible Reach SIP Trunk. All calls received from Source IP Group 2 (i.e., from Lync 2010 Mediation Server) and the destination number is '1511', remove the '1' prefix from the left.
- Index #5 defines destination number manipulation of IP calls from AT&T IP Flexible Reach SIP Trunk. All calls received from Source IP Group 2 (i.e., from Lync 2010 Mediation Server) and the destination number is '1911', remove the '1' prefix prefix from the left.

5.9.2 Configure Source Phone Number Manipulation

- To configure Source Phone Number Manipulation Table for IP -> Tel Calls Table:
 - Open the 'Source Phone Number Manipulation Table for IP -> Tel calls' page (Configuration tab > VoIP menu > GW and IP to IP > Manipulations submenu > Source Number IP >Tel).

ndex	Destination Prefix	Source Prefix	Source IP Addre	ss Stripped Digits From Left	Stripped Digits From Right	l Prefix t	o Add	Suff	fix to Add	Nur
	•	+1	10.15.9.11	2	0					255
0	•	+	10.15.9.11	1	0					255
0	÷	1	10.15.9.11	1	0					255
0	•	anonymous	10.15.9.11	20	0	a7192083390				255
				(
			, Number of Digits to Leave	NPI	7	TON	Presei	ntation		
		2	Number of Digits to Leave	NPI Not Configured	Not	TON	Preser	ntation		
		2	Number of Digits to Leave	NPI Not Configured	Not	TON t Configured t Configured	Preser Allowed Allowed	ntation		
		2 2 2	Number of Digits to Leave	NPI Not Configured Not Configured Not Configured	Not	TON t Configured t Configured t Configured	Preser Allowed Allowed Allowed	ntation		

Figure 5-18: Source Phone Number Manipulation Table for IP -> Tel Calls Page

- Index #1 defines Source number manipulation of IP calls from Lync Mediation Server. All calls received from IP address 10.15.9.11 (i.e., from Lync Mediation Server) and the source number prefix begins with '+1', remove the '+1' from the number.
- Index #2 defines Source number manipulation of IP calls from Lync Mediation Server. All calls received from IP address 10.15.9.11 (i.e., from Lync Mediation Server) and the source number prefix begins with '+', remove the '+' from the number.
- Index #3 defines Source number manipulation of IP calls from Lync Mediation Server. All calls received from IP address 10.15.9.11 (i.e., from Lync Mediation Server) and the source number prefix begins with '1', remove the '1' from the number.
- Index #4 defines Source number manipulation of anonymous calls from Lync Mediation Server. Anonymous calls received from IP address 10.15.9.11 (i.e., from Lync Mediation Server) replace the 'anonymous' caller ID with a modified well known number i.e. a7192083390. This manipulation is performed to create a well known number in the P-Asserted-Identity header. Without this number, the AT&T IP Flexible Reach SIP Trunk rejects the call. See later on for the Source Number manipulation Tel->IP manipulation rule that restricts the caller ID for an anonymous call on page 29 within Source Phone Number Manipulation Table for Tel -> IP index #4.

To configure Source Phone Number Manipulation Table for Tel -> IP Calls Table:

 O pen the 'Source Phone Number Manipulation Table for Tel -> IP calls' page (Configuration tab > VoIP menu > GW and IP to IP > Manipulations submenu > Source Number Tel > IP).

Figure 5-19: Source Phone Number Manipulation Table for Tel -> IP Calls Page

Ind	lex	Source Trunk Group	Source IP Group	Destination Prefix	Source Prefix	Stripped Digits From Left	Stripped Digits From Right	Prefix to Add	Suffix to Add	Number of Digit Leave
0	0	-1	1	*	+	0	0			255
1	0	-1	1	+	1	0	0	+		255
2	0	-1	1	*	Restricted	0	0			255
з	0	-1	1	•	XXXXXXXXXXX#	0	0	+1		255
4	0	-1	2	*	a7192083390	1	0			255
								Presentation Allowed Allowed Not Configured Allowed Restricted		

- Index #0 defines Source number manipulation of IP calls from AT&T IP Flexible Reach SIP Trunk. All calls received from Source IP Group 1 (i.e., from AT&T IP Flexible Reach SIP Trunk) and the Source number prefix begins with '+', do not perform any changes to the number.
- Index #1 defines Source number manipulation of IP calls from AT&T IP Flexible Reach SIP Trunk. All calls received from Source IP Group 1 (i.e., from AT&T IP Flexible Reach SIP Trunk) and the Source number prefix begins with '1', Add a '+' as a prefix to the number.
- Index #2 defines Source number manipulation of IP calls from AT&T IP Flexible Reach SIP Trunk. All calls received from Source IP Group 1 (i.e., from AT&T IP Flexible Reach SIP Trunk) and the Source number prefix is set as 'Restricted', do not perform any changes to the call.
- Index #3 defines Source number manipulation of IP calls from AT&T IP Flexible Reach SIP Trunk. All calls received from Source IP Group 1 (i.e., from AT&T IP Flexible Reach SIP Trunk) and the Source number length is 10 digit number, add the '+1' prefix to the number.
- Index #4 defines Source number manipulation of anonymous calls from Microsoft Lync environment. All calls received from Source IP Group 2 (and the Source number was modified to a7192083390 (which is a specially modified well known number that was inserted for the anonymous caller ID on the source number manipulation IP->Tel above), the 'a' is removed and the presentation should be set to 'restricted'. To simply mark all calls as Private calls, use an asterisk '*' in the Source Prefix field. Individual Telephone Numbers or ranges can also be set in this manner as well.

5.10 Step 11: Configure IP Profile for Call Forwarding

One of the challenges with the integration of the Microsoft Lync 2010 server and the Flexible Reach SIP Trunk is the implementation of call forwarding. Since the Microsoft Lync client forwards the call back to the SIP Trunk, it does not provide any information in the forwarded INVITE (such as Diversion header) informing that this call has been forwarded. Consequently, it is necessary to configure a special IP Profile that adds the diversion header toward the SIP trunk in the event of a call forwarding scenario.

This profile is later associated to the routing table in the event of a call forwarding scenario (see section 5.8.1 on page 49).

To configure IP Profile for call forwarding:

 Open the 'IP Profile Settings' page (Configuration tab > VoIP menu > Coders and Profiles > IP Profile Settings).

▼			
Profile ID	(2)→	3	~
Profile Name		Lync Transfers	
 Common Parameters 			
- Gateway Parameters			
Fax Signaling Method		No Fax	~
Play Ringback Tone to IP		Don't Play	~
Enable Early Media	~	Enable	*
Copy Destination Number to Redirect Number		Before Manipulation	~
Media Security Behavior		Disable	~
CNG Detector Mode		Disable	*
Modems Transport Type		Enable Bypass	*
NSE Mode		Disable	*
Number of Calls Limit		-1	
Progress Indicator to IP		Not Configured	*
Profile Preference	~	1	*
Coder Group	(4)→	Default Coder Group	*

Figure 5-20: IP Profile Settings for Call Forwarding "numbers"

- 2. From the 'Profile ID' drop-down list, select '3'.
- **3.** From the 'Copy Destination Number to Redirect Number' drop-down list, select 'Before Manipulation'; this parameter adds the Diversion Header to the INVITE in event of a call forwarding scenario.
- 4. From the 'Coder Group' drop-down list, select 'Default Coder Group'.

- Open the 'Admin" page, by appending the case-sensitive suffix 'AdminPage' to the Media Gateway's IP address in your Web browser's URL field (e.g., http://10.15.4.15/AdminPage).
- 6. On the left pane, click *ini* Parameters.

Figure 5-21: Output Window

Image Load to Device	Parameter Name: Enter Value: USESIPURIFORDIVERSIONHEADER	Apply New Value
<i>ini</i> Parameters	Output Window	
Back to Main	Parameter Name: USESIPURIFORDIVERSIONHEADER The Value is invalid: Parameter Current Value: 1 Parameter Description:Use Tel uri or Sip uri for Diversion header	•

- 7. In the 'Parameter Name' field, enter the parameter USESIPURIFORDIVERSIONHEADER. In the 'Enter Value' field, enter "1".
- 8. Click Apply New Value.

5.10.1 Configure Redirect Number Manipulation

In the event of a call forwarding scenario, a Diversion header needs to be added to the INVITE towards the Flexible Reach SIP Trunk (as configured in Section 5.10 above). In this case, the E-SBC copies the Destination number to the Redirect number and adds this number to the Diversion header. In order to have a well known number in the Diversion header (for Flexible Reach SIP Trunk), a manipulation rule should be defined to replace the redirect number to a well known number.

To configure redirect number Tel -> IP Table:

 Open the 'Redirect Number Tel -> IP' page (Configuration tab > VoIP menu > GW and IP to IP > Manipulations submenu > Redirect Number Tel > IP).

Figure 5-22: Redirect Number Tel -> IP Page

Ind	lex	Source Trunk Group	Source IP Group	Destination Prefix	Redirect Prefix	Stripped Digits From Left	Stripped Digits From Right	Prefix to Add	Suffix to Add	Number of Digit Leave
1	0	-1	-1	•	*	20	0	7323680193		255

Index #1 defines redirect number manipulation for the call forwarding scenario.

The redirect number is changed to a well known number i.e. 7323680193.

5.11 Step 12: Configuring SIP General Parameters

This section describes how to configure the SIP general parameters.

> To configure general SIP parameters:

 Open the 'SIP General Parameters' page (Configuration tab > VoIP menu > SIP Definitions submenu > General Parameters).

•	SIP General			
4	NAT IP Address	2		
	PRACK Mode		Supported	*
	Channel Select Mode		Cyclic Ascending	*
	Enable Early Media	3→	Enable	*
	183 Message Behavior		Progress	*
	Session-Expires Time		0	
	Minimum Session-Expires		90	
	Session Expires Method		Re-INVITE	¥
	Asserted Identity Mode	4→	Adding PAsserted Identity	¥
	Fax Signaling Method		No Fax	*
	Detect Fax on Answer Tone		Initiate T.38 on Preamble	*
	SIP Transport Type	5 →	TCP	¥
	SIP UDP Local Port		5060	
	SIP TCP Local Port	6 →	5060	
	SIP TLS Local Port	Ŭ	5061	
	Enable SIPS		Disable	*
	Enable TCP Connection Reuse		Enable	~
	TCP Timeout		0	
	SIP Destination Port		5060	
	Use user=phone in SIP URL		Yes	*
	Use user=phone in From Header		No	*
	Use Tel URI for Asserted Identity		Disable	¥
	Tel to IP No Answer Timeout		180	
	Enable Remote Party ID		Disable	*
	Add Number Plan and Type to RPI Header		Yes	*
	Enable History-Info Header		Disable	¥
	Use Source Number as Display Name		No	*
	Use Display Name as Source Number		No	*
	Enable Contact Restriction		Disable	Y

Figure 5-23: SIP General Parameters Page

	Enable Contact Restriction	Disable	*
	Play Ringback Tone to IP	Don't Play	~
	Play Ringback Tone to Tel (7)	Play Local Until Remote Media A	¥
	Use Tgrp information	Disable	¥
	Enable GRUU	Disable	¥
	User-Agent Information		
	SDP Session Owner	AudiocodesGW	
	Play Busy Tone to Tel	Don't Play	¥
	Subject		
	Multiple Packetization Time Format	None	*
	Enable Semi-Attended Transfer	Disable	~
	3xx Behavior	Forward	¥
	Enable P-Charging Vector	Disable	¥
	Enable VoiceMail URI	Disable	*
	Retry-After Time	0	
	Enable P-Associated-URI Header	Disable	¥
	Source Number Preference		
	Forking Handling Mode 8	Sequential handling	¥
	Enable Comfort Tone	Enable	~
	Add Trunk Group ID as Prefix to Source	No	~
	Fake Retry After (9)	60	
	Enable Reason Header	Enable	~
_			

- 2. In the 'NAT IP Address' field, ensure this field is empty. It is not used for the Single LAN Interface solution. This field is only used when the WAN Interface is utilized with a Global (public) IP address of the E-SBC device to enable the static NAT between the E-SBC device and the Internet.
- 3. From the 'Enable Early Media' drop-down list, select 'Enable' to enable early media.
- 4. From the 'Asserted Identity Mode' drop-down list, select 'Adding PAsserted Identity'.
- 5. From the 'SIP Transport Type' drop-down list, select 'TCP' in case the Mediation Server is configured to use TCP transport Type.
- 6. In the 'SIP TCP Local Port' field, enter '5060'; this port is the listening E-SBC device port for TCP transport type. This port must match the transmitting port of the Mediation Server.
- 7. From 'Play Ringback Tone to Tel' drop-down list, select 'Play Local Until Remote Media Arrive'. Plays the RBT according to the received media. If a SIP 180 response is received and the voice channel is already open (due to a previous 183 early media response or due to an SDP in the current 180 response), the E-SBC device plays a local RBT if there are no prior received RTP packets. The E-SBC device stops playing the local RBT as soon as it starts receiving RTP packets. At this stage, if the E-SBC device receives additional 18x responses, it does not resume playing the local RBT.
- 8. From the 'Forking Handling Mode' drop-down list, select 'Sequential handling'; this parameter determines whether18x with SDP is received. In this case, the E-SBC device opens a voice stream according to the received SDP. The E-SBC device re-opens the stream according to subsequently received 18x responses with SDP.



Reasons for

9. In the 'Fake Retry After' field, enter '60' sec. This parameter determines whether the E-SBC device, upon receipt of a SIP 503 response without a Retry-After header, behaves as if the 503 response included a Retry-After header and with the period (in seconds) specified by this parameter.

5.12 Step 13: Defining Reasons for Alternative Routing

A 503 SIP response from the Mediation Server to an INVITE must cause the E-SBC device to perform a failover. In other words, if the Lync Mediation Server primary proxy server is not responding, an attempt is made to establish communication with the secondary proxy server. For this event to occur, you need to perform the following actions:

- Configure the Reasons for Alternative Routing for Tel-to-IP calls to '503 SIP response'.
- Configure the Lync Mediation Proxy Set for redundancy purposes.

> To define SIP Reason for Alternative Routing:

 Open the 'Reasons for Alternative Routing' page (Configuration tab > VoIP menu > GW and IP to IP > Routing submenu > Alternative Routing Reasons).

IP to Tel Reasons			
Reason 1		~	
Reason 2		~	
Reason 3		~	
Reason 4		×	
Tel to IP Reasons			
Reason 1	503	⊻ ← _2	
Reason 2		~	
Reason 3		~	
Reason 4		~	

Figure 5-24: Reasons for Alternative Routing Page

- 2. Under the Tel to IP Reasons group, for Reason 1, select '503'.
- 3. Click Submit.
- 4. Open the 'Proxy & Registration' page (Configuration > VoIP > SIP Definitions > Proxy & Registration) and configure the 'Redundant Routing Mode' parameter to 'Proxy' as shown below in Figure 5-25. This will allow entry back into the Proxy Set table for the next available route. This redundant route is configured in the next step (on Proxy Set ID 2, see Figure 5-26 below).

 Open the 'Proxy Sets Table' page (Configuration tab > VolP menu > Control Network > Proxy Sets Table). Configure the Proxy Set for the Lync Mediation Server:

From the 'Proxy Set ID' drop-down list, select "2".

- **a.** In the 'Proxy Address' column, enter a second IP address or the FQDN and the listening port of the secondary Lync Mediation Server.
- b. From the 'Is Proxy Hot Swap' drop-down list, select "Yes".
- 6. Click Submit.

▼	
Use Default Proxy	No 👻
Proxy Name	
Redundancy Mode	Parking -
Proxy IP List Refresh Time	60
Enable Fallback to Routing Table	Disable 👻
Prefer Routing Table	No 👻
Always Use Proxy	Disable 👻
Redundant Routing Mode	Proxy - 4
SIP ReRouting Mode	Standard Mode 👻
Enable Registration	Disable 👻
Registration Time	180
Re-registration Timing [%]	50
Registration Retry Time	30
Registration Time Threshold	0
Re-register On INVITE Failure	Disable 👻
ReRegister On Connection Failure	Disable 👻

Figure 5-25: 'Proxy & Registration' Page



2 Proxy Set ID ¥ Proxy Address Transport Type 10.15.9.11:5060 тср 🗸 1 TCP 🔽 < (5a) 2 10.15.9.12:5060 3 ¥ 4 Y 5 ¥ Enable Proxy Keep Alive Using Options ¥ Proxy Keep Alive Time 60 Proxy Load Balancing Method Disable ¥ Is Proxy Hot Swap (5b) Yes ¥ Proxy Redundancy Mode Not Configured ~

Figure 5-26: Proxy Set ID 2 for Lync Mediation Server

6 Troubleshooting

This section should provide some tips for troubleshooting problems, including troubleshooting commands and contact numbers within Vendor X's Company for trouble escalation.

6.1 Debugging Procedures

This section discusses the following debugging procedures:

- Case Reporting Procedures. See section 6.1.1 below.
- Syslog. See section 6.1.2 on page 66.
- Wireshark Network Sniffer. See section 6.1.3 on page 68.

6.1.1 Case Reporting Procedures

When reporting a problem to AudioCodes' Technical Support department, the following information should be provided:

- Basic information (required for all types of problems):
 - Problem description (nature of failure, symptoms, call direction, etc.)
 - Network diagram
 - *ini* configuration file (downloaded to your PC from the device using the Web interface)
 - Syslog trace (without missing messages)
 - Unfiltered IP network trace using the Wireshark application

(Note: If you are unable to collect all the network traffic, then at least collect the mandatory protocols SIP, RTP, and T38.)

- Advanced information (if required upon request):
 - PSTN message traces for PSTN problems
 - Media stream traces for problems related to voice quality, modem\fax, DTMF detection, etc.

6.1.2 Syslog

Syslog is a standard for forwarding log messages in an IP network. A syslog client, embedded in the device sends error reports/events generated by the device to a remote Syslog server using IP/UDP protocol. This information is a collection of error, warning and system messages that record every internal operation of the device. You can use the supplied AudioCodes proprietary Syslog server "ACSyslog" (shown in the figure below) or any other third-party Syslog server for receiving Syslog messages.

🎒 ACSyslog ;-)	R0.9.9		×
File View Search	Options Help		
- 🔊 🕏	ا 👟 🔬	$\mathbf{\overline{O}}$	
🕑 Time	📃 Hast	🔯 Priority	Message
13:29:29.510	10.13.4.13	WARNING	(lgr_psbrdif)(20) !! [ERROR] #1:failed to play to
13:29:29.510	10.13.4.13	WARNING	Invalid Tone Type (7). Channel ID:1 [Code:500c][CID:
13:29:21.949	10.13.4.13	NOTICE	(lgr_endpoint)(19) FXSEndPoint::HandleDialedStr
13:29:21.949	10.13.4.13	NOTICE	(lgr_digitmap_mngr)(18) DigitMapMngr::HandleDia
13:29:17.200	10.13.4.13	WARNING	(lgr_psbrdif)(17) !! [ERROR] #1:failed to play tor
13:29:17.200	10.13.4.13	WARNING	Invalid Tone Type (7). Channel ID:1 [Code:500c][CID:
<			>
0 message(s) per min	ute (6/26)	0.9.9/Unkno	own 6 6 0 -

Figure 6-1: AudioCodes' Proprietary Syslog Server

> To activate the Syslog client on the device using the Web interface:

- Open the 'Syslog Settings' page (Configuration tab > System menu > Syslog Settings).
- 2. In the 'Syslog Server IP Address' field, enter the IP address of the Syslog server (*ini* file parameter SyslogServerIP).
- **3.** From the 'Enable Syslog' drop-down list, select 'Enable' to enable the device to send syslog messages to a Syslog server (defined in Step 2).

Figure 6-2: Enabling Syslog

Enable Syslog	Disable	*
Syslog Server IP Address		
Syslog Server Port	514	
Debug Level	0	*
Analog Ports Filter	-1	
Trunks Ports Filter	-1	

4. From the 'Debug Level' drop-down list, select '5' if debug traces are required. To enable syslog reporting, using the *ini* file, load an *ini* file to the device with the following settings:

```
[Syslog]
SyslogServerIP = 192.168.2.35
EnableSyslog = 1
SyslogServerPort = 514
GWDebugLevel = 5
```

6.1.3 Wireshark Network Sniffer

Wireshark is a freeware packet sniffer application that allows you to view the traffic that is being passed over the network. Wireshark can be used to analyze any network packets. Wireshark can also be used to analyze RTP data streams and extract the audio from the data packets (only for G.711). The audio can be saved as a *.pcm file.

To record traffic that is sent to / from the device:

- 1. Install Wireshark on your PC. (You can download it from the following Web site: http://www.wireshark.org/
- 2. Connect the PC and the device to the same hub.
- **3.** If you are using a switch, use a switch with port mirroring for the port to which the Wireshark is connected.
- 4. Start Wireshark.
- Select the network interface that is currently being used by the PC on the toolbar,click Interfaces, and then in the 'Capture Interfaces' dialog box, click the Options button corresponding to the network interface:

Figure 6-3: Selecting Interface Currently used by the PC

	go Capture	Analyze	Scatistics	Heb	13	x	2	A	10	4	3
nterfaces Optio	ns Start	Stop	Restart	Open	Save As	Close	Refres	i Park	And	Back	,
iter List the avail	able capture inte	rfaces				• <u>E</u> xp	ression	Clear Apply	e e		
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terfaces								17			
	1000	De	iscription		1P	Packets	Packets/s		402		
	Mapter	for generic	: dalup and V	PN capture	unknown			Start Op	ions Details		
	Broadco	m Net30trem	ve Gigabit Eth	ernet Driver	10.13.22.6	114		Sart Op	ions Details		
	Help								⊈lose		

6. In the 'Capture Options' dialog box, select the desired display options:

Figure 6-4: Configuring Wireshark Display Options

Capture	
Interface: Broadcom NetXtreme Gigabit Ethernet Driver:	: \Device\NPF_{20F728EA-2789-45E1-8
IP address: 10.13.22.6	
tink-liver header type: Ethernot 🗸 Buffer ster: 1	megabyte(s) Wreless Settings
Capture paciets in promiscuous mode	
Unit each packet to 💷 🗧 bytes	
Gapture Filter:	
Capture File(s)	Display Options
File: Browse	🛄 🔽 Update list of packets in real time
Use multiple files	
Next file avery 1 negabyte(s)	Automatic scrolling in live capture
i (lest file every). C menute(s)	🛛 🗹 Hide capture info dialog
Ring butter with 2 0 tiles	Alexa Principalina
Dop cature after 1 Divisi	nane resolution
Rop Capture	Image: Image
after 1 puckat(s)	Enable getwork name resolution
after I megaliyte(s)	
after 1 1 Herute(c)	Enable transport name resolution



7. Click Start.

Figure 6-5: Captures Packets

Record Com HetXt	çone Gigabii	Ethernet	Driver: Cap sica geu	pturing Wi	reshark						
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tet sp		-	_	Filter	Bar	ion. De	ar 6001				
o Time	5	N/DR	L	Destrution		Protocol	2rfe .				
163 09:35:2 164 09:35:2 209 09:35:2 210 09:35:2	2.327114 1 2.338134 1 2.362899 1 2.393918 1	0,33,6,1 0,33,6,1 0,33,6,1 0,33,6,1	01	Packet li	ist pane	SIP SIP SIP SIP	Status Status Request Status	100 Try 180 R1r PRACK 200 OK	1ng 1ng 1p:201010), 33. 6. 10	01
Frame 152 (80 Ethernet II,	STC: AUd1	n wire. ococ_oa:	861 byte 8a:52 (0	s capture 0:90:8f:0	d) a:6a:b2),	DSE: AU	d10c.0d_0	a:70:00 (00:90:6f:0	041761¢¢)
Frame 152 (M Ethernet II, Internet Prot User Datagram Session Init	SI bytes of src: audi cocol, src Protocol ation Pro	n wire, ococ_oa: : 10.33. , Src Po tocol	861 byte sa:52 (0 6.100 (1 rt: 5060	s capture 0:90:8f:0 0.33.6.10 (1060),	d) a:8a:b2), 0), Ost: Dst Port:	DST: AU 10.33.6. 5060 (5	diocod_0 101 (10. 060)	a:7c:cc (33.6.101)	00:90:67:0	04:76:66)
Frame 122 (B) Ethernet II. Internet Prot User Datagram Session Init: USSION FRI Via: SIP/2. Max-Forward From: csip To: csip:20 Call-1D: 5 Cfec: 1 Two	51 bytes of Src: audi cocol, Src Protocol lation Pro 201010, 33 0/UDP 10. 55: 70/r/r 101010, 33, 6 1956707171 (TE)/r/r	n wire, ocod_0a: : 10.33, , Src Po tocol : 6.101; 33.6.100 : 6.100>; .101; use 20002349	661 byte sa:b2 (0 6.100 (1 rt: 5060 pr:	s capture 0:90:8f:0 0,33.6.10 (1060), w text) e sIP/2.0 Packet de r/n .6.100/r	d) a:Sa:b2), 0), 0ST: DST Port: Nr\n tails pand	DSE: AU 10.33.6. 5060 (5	d1ocod_0 101 (10. 060)	a:7c:cc (33.6.101)	00:90:6f:(04176166	2
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Frame 122 (b) Sthernet IX. Internet Prot USer Datagram Session Init SUSSIONE OFF INITE SIP/ Nax-Forwar From: <sip To: <sip:1 <br="">Call-DD: & CSeq: 1 IW Contact: :: Connertad: 0 66 65 11</sip:1></sip 	SI bytes c src: audi ocol, Src Protocol ation Pro 201010.33 0.00P 10. 55: 70 r/r 101010.33.6 956707171 0101010 es 100rel 101010 es 100 es 1	n wire, ococ_oa: : 10.33, . Src Po tocol 33.6.101: . 000: . 0.101: . 0.100: . 0.100:	861 byte sa:b2 (0 6.100 (1 rt: 5060 PISTE ser-phone tag-1c r-phones 25010.33 0>\r\n b1 tag-1c b3 tag-1c	s capture 0:90:81:00 (1060), a toxt) e s1P/2.0 Packet de r.n .6.100 r	d) a:sa:b2), 0), 0st: Dst Fort: Vr\n tails pand n	DSE: AU 10.33.6. 5060 (5	discot_0 101 (10. 060)	a:7c:cc (33.6.101)	00:#0:8f:(OA:7C:CC	2
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Frame 132 (4) Ethernet 12. Internet Provides Datagram Session Init Session Init INVITE Sip Via: SIP/2. Max-Formair From: <sip To: <sip:24 Call-DD: B- CSeq: 1 IW Contact: co Computed 13 Co 66 51 11 Co 66 51 12 Co 66 51 12 Co 66 51 12 Co 65 12 Co 65</sip:24 </sip 	1 bytes c src: audi ocol. Src Protocol ation Protocol 0.000 10. 5: 70'r'e 101010.33. 0.05670171 101010.33. 0.95670171 0.1010.33. 0.95670171 0.1010.33. 0.95670173 0.95670173 0.95670173 0.95770173700000000000000000000000000000000	n wire, ococoa: 10.33, 5rc Po tocol 6.101: 33.6.100 101:0se 20002349 .33.6.10 .101:0se 20002349 .33.6.10 .101:0se 20002349 .33.6.10 .101:0se .001:0se	861 byte 8a:b2 (0 6.100 (1 rt: 5000 10 15 r ser-phone 25010, 33 0>\r\n b3 15 25010, 33 0>\r\n b3 15 25010, 33 0>\r\n	s capture o societ co o soci o societ co o societ co o societ co o societ co o	d) arsa:b2), 0), ost: Dst Port: Dst Port: tails pand n n tails pand n tails pand	Dit : AU 10.33.6, 5060 (5 7 m 1 m 1 m 1 m 1 m 1 m 1 m 1 m 1 m 2 m 1 m 2 m 1 m	d1ocod_0 101 (10, 060) 106 316 shone v1a1 51 10, 315 v10, 315	a:70:00 (00:#0:df:(04176166	2

8. To view VoIP call flows, from the **Statistics** menu, choose **VoIP Calls**. You can view the statistics in graph format by clicking **Graph**.

Figure 6-6: Viewing VoIP Call Flows

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9. To play G.711 RTP streams, click the **Player** button.

Figure 6-7: Playing G.711 RTP Streams



10. To analyze the RTP data stream and extract the audio (which can be played using programs such as CoolEdit) from the data packets (only for G.711), from the **Statistics** menu, point to **RTP**, and then choose **Stream Analysis**.



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- a. Save the audio payload of the RTP stream to a file.
- **b.** Save the Payload as a *.pcm file.
- c. Select the 'forward' option.

6.2 Verifying Firmware

To verify the firmware load actively running on the device, log into the device and view the firmware version on the product homepage as shown in the figure below.

Figure 6-9: Viewing active firmware version

Mediant	1000 - MSBG Submit 🙆 Burn Device Actions 🔹 👘 Hone 🔞 Help 🐑 Log off
Configuration Meintenance Status & Diagnostics Search	Mediant 1000 - MSBG Home Page
© Basic © Full © © System © OrP © Data	1 2 Digital 3 CRMx Image: Comparison of the second
	General Information Trunk (Digital Modules) P Address 10.15.9.118 Subnet Mask 255.255.00 Defaul Gateway 10.15.9.117 Digta Fork Number 1 Firmware Version 6.204.020.002 Protocol Type SP Gateway Operational State UNLOCKED
Reader's Notes



Mediant 800 MSBG E-SBC, Mediant 1000 MSBG E-SBC and Mediant 3000 E-SBC Media Gateway

Configuration Note

Connecting AT&T's IP Flexible Reach - MIS/PNT/AVPN SIP Trunking Service

to

Microsoft® Lync Server 2010

Using

AudioCodes' Mediant 800 E-SBC, Mediant 1000 MSBG E-SBC and Mediant 3000 E-SBC Media Gateway

Document #: LTRT-38100



www.audiocodes.com